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EXAMINER	
HALIYUR, VENKATESH N	

ART UNIT	PAPER NUMBER
2419	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/822,447	KUBLER ET AL.	
	Examiner	**Art Unit**	
	VENKATESH HALIYUR	2419	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 02 February 2009.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-59 (claims 2,23 canceled)* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☒ Claim(s) *43-52* is/are allowed.

6)☒ Claim(s) *1,3,5-7,9-16,18-22,24,26-28,30-37,39-42,53 and 55-59* is/are rejected.

7)☒ Claim(s) *4,8,17,25,29,38 and 54* is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *08 April 2004* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. The amendments filed on 02/02/2009 have been fully considered. However new ground(s) of rejections have been made in this office action and therefore the rejection communicated via previous office action has been withdrawn. Rejection follows.

2. Claims 1-59 are pending in the application. Claims 2, 23 canceled. Claims 43-59 are new. The terminal disclaimer 09/04/2008 for USPAT: 5,726,984 has been approved and recorded

Claim Rejections – 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 1,3,5-7,9-16,18-22,24,26-28,30-37,39-42 are rejected under 35 U.S.C. 103(a) as being unpatentable over Huang [US Pat: 5,434,856] in view of Oneill et al [US Pat: 5,987,099] further in view of Harrison et al [US Pat 5,796,727].

Regarding claims 1,22 Huang in the invention of "Method for Monitoring Communication Talkgroups" disclosed a communication network **(Figs 1-3)** operating to support voice and data communication within a premises, said communication network comprising: a plurality of mobile network devices **(items 114-116, of Fig 1)** comprising a buffer **(item 200 of Fig 2)** that stores incoming digital voice information for a predetermined queuing period before beginning voice reproduction from the stored digital voice information **(col 1, lines 58-67)**; a stationary network device **(packet gateway, item 121 of Fig 1)**; a wireless network **(item 111 of Fig 1)** that is used by each of said plurality of mobile network devices **(items 101-103 of Fig 1)** to selectively exchange voice and data packets with others of the plurality of mobile network devices; a hardwired network **(LAN, item 125 of Fig 1)** connected to both said stationary network device **(packet gateway)** and said wireless network **(items 111-113 of Fig 1, col 2, lines 1-23)**; said hardwired network being used to route voice and data packets between said stationary network device and said plurality of mobile network devices which participate via said wireless network **(col 1, lines 58-67)**; a telephone **(consoles, item 122 of Fig 1)**, connected to said stationary network device, that captures, delivers, receives and reproduces voice in an analog voice stream form **(col 2, lines 41-52);** said stationary network device comprising a buffer that stores digital voice information received from said wireless network for a predetermined queuing period before converting the stored digital voice information **(voice packets)** into an analog voice stream **(D/A, item 210 of Fig 2, col 3, lines 1-42)** and delivering the analog voice

stream to said telephone **(col 2, lines 24-40, lines 53-67)** but fails to disclose said stationary network device converts analog voice streams received from said telephone into voice packets for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices. However, Oneill et al in the invention of “Low-Power Wireless System for Telephone Services” disclosed a method for said stationary network device **(base station controller, item 618 of Fig 5)** that converts analog voice streams received from said telephone into voice packets for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices **(col 5, lines 56-65, Fig 5).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method of converting analog voice streams received from said telephone into voice packets at the stationary device for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices as taught by Oneill et al in the system of Huang for the stationary network device to converts analog voice streams received from said telephone into voice packets for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices.

Huang and Oneill fail to positively disclose the feature of buffering the digital voice information received from said wireless network for a predetermined queuing period before converting the stored digital voice information into an analog voice stream and delivering the analog voice stream to said telephone.

However, Harrison et al disclosed a communication network **(Figs 2-3, col 3, lines 5-31, col 5, lines 8-40)**, wherein the voice packets are queued at LAN network and managed by a stream manager corresponding to the latency of the network before being sent to the MS **(mobile stations, col 16, lines 7-20).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method for queuing voice packets corresponding to the latency of the network before being sent to the MS as taught by Harrison et al in the system of Huang as modified by Oneill et al for including the feature of buffering the digital voice information received from said wireless network for a predetermined queuing period before converting the stored digital voice information into an analog voice stream and delivering the analog voice stream to said telephone. One is motivated as such in order to convert analog voice streams to digital voice packets to efficiently route packets over the voice packet network for deliver to mobile network devices.

Regarding claims 3,19,24,40, Huang disclosed said stationary network device is a computer **(routers, col 2, lines 53-67)**.

Regarding claim 5, 26, 32 Huang disclosed that said stationary network device provides call setup assistance for said telephone **(communication links established via base stations and gateway with communication units, col 2, lines 3-13)**.

Regarding claims 6,16,27,37, Huang disclosed a telephone switching network **(frame relay switch, item 120 of Fig 1)** connected to said stationary network device

(packet gateways and routers, item 121 of Fig 1); and said stationary network device selectively routes analog voice streams received from said telephone onto said telephone switching network, and said stationary network device selectively routes analog voice streams received from said telephone switching network to said telephone **(col 2, lines 4-40)**.

Regarding claim 7,28, Huang disclosed a communication network located within a premises for supporting voice and data exchanges **(Figs 1-3),** said communication network comprising: a plurality of portable terminals **(items 101-103 of Fig 1)**, each comprising a wireless transceiver; each of said plurality of portable terminals capture voice in an analog voice stream form and generate therefrom digital voice packets **(col 2, lines 3-24)**, and each of said plurality of portable terminals **(items 101-103, of Fig 1)** receive digital voice packets, generate therefrom analog voice streams, and reproduce voice from the analog voice streams **(col 3, lines 51-65);** each of said plurality of portable terminals capture data and generate therefrom data packets, and each of said plurality of portable terminals receive data packets and reproduce data from the data packets received **(col 3, lines 3-10)**; a plurality of access devices **(base stations, items 114-116 of Fig 1)**, each comprising a wireless transceiver **(item 22 of Fig 1).**

Huang disclosed portable terminals and routing data/voice packets to portable terminals **(col 2, lines 23-40),** but fails to disclose said plurality of access devices using a polling protocol to manage wireless routing of data and voice packets within the premises among said plurality of portable terminals and mobile stations

comprising wireless transceiver communicating with wireless network and stationary networks including data networks. However, Oneill et al disclosed a method for plurality of access devices **(base stations)** using a polling protocol to manage wireless routing of data and voice packets within the premises among said plurality of portable terminals **(col 6, lines 48-62, col 12, lines 45-51, Figs 3,10).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method for plurality of access devices using a polling protocol to manage wireless routing of data and voice packets within the premises among said plurality of portable terminals as taught by Oneill et al in the system of Huang for plurality of access devices to use a polling protocol to manage wireless routing of data and voice packets within the premises among said plurality of portable terminals.

Huang and Oneill fail to positively disclose the feature of buffering the digital voice information received from said wireless network for a predetermined queuing period before converting the stored digital voice information into an analog voice stream and delivering the analog voice stream to said telephone.

However, Harrison et al disclosed a communication network **(Figs 2-3, col 3, lines 5-31, col 5, lines 8-40)**, wherein the voice packets are queued at LAN network and managed by a stream manager corresponding to the latency of the network before being sent to the MS **(mobile stations, col 16, lines 7-20).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method for queuing voice packets

corresponding to the latency of the network before being sent to the MS as taught by Harrison et al in the system of Huang as modified by Oneill et al for including the feature of buffering the digital voice information received from said wireless network for a predetermined queuing period before converting the stored digital voice information into an analog voice stream and delivering the analog voice stream to said telephone. One is motivated as such in order to use a polling protocol to efficient and fair service for data transfers to/from the mobile terminals with the access devices.

Regarding claims 9,13,30,34 Huang disclosed that a telephone, connected to one of said plurality of access devices **(base stations),** that captures, delivers, receives and reproduces voice in an analog voice stream form **(col 2, lines 41-52)**; said one of said plurality of access devices selectively converting digital voice packets received into an analog voice stream for delivery to said telephone for reproduction **(col 3, lines 3-11)**; and said one of said plurality of access devices selectively converting an analog voice stream received from said telephone into digital voice packets for delivery to one of said plurality of portable terminals **(col 3, lines 12-65).**

Regarding claim 10, 31 Huang disclosed a telephone switching network **(frame relay switch)** connected to said one of said plurality of access devices **(base stations)**; said one of said plurality of access devices selectively routes analog voice streams received from said telephone through said telephone switching network; and said one of said plurality of access devices selectively routes analog voice

streams received from said telephone switching network to said telephone **(col 2, lines 23-40).**

Regarding claims 11, 14, 35, Huang disclosed that said one of said plurality of access devices provides call setup assistance for said telephone **(communication links established via base stations with communication units, col 2, lines 3-13)**.

Regarding claim 12, 33 Huang disclosed one of said access devices stores incoming digital voice packets for a queuing time period before converting the digital voice packets into an analog voice stream form **(D/A, item 210 of Fig 2, col 2, lines 53-67, col 3, lines 1-42).**

Regarding claims 15,36 Huang et al disclosed A communication network **(Fig 1)** for supporting voice exchanges, said communication network comprising: a voice stream network **(voice packet pertaining to particular communication talk group)** that selectively routes voice signals captured in an analog voice stream form **(voice packets decoded and transmitted, Fig 1, col 1, lines 55-67, col 2, lines 4-13);** a voice packet network, independent of said voice stream network, that selectively routes voice in a digital voice packet form **(col 2, lines 14-22)**; a first network device **(base station, item 114 of Fig 1)** that captures and delivers voice in the analog voice stream form, and said first network device receives and reproduces voice from the analog voice stream form **(col 3, lines 3-11)**; a second network device **(frame relay, item 120 of Fig 1)** independent of said first network device, that communicatively couples with said first network device to receive and deliver

voice in the analog voice stream form **(base station and frame relay connected via communication links);** said second network device selectively interfaces with said voice stream network to receive and route voice for said first network device in the analog voice stream form **(col 2, lines 24-29);** said second network device selectively interfaces with said voice packet network to receive and route voice for said first network device in the digital voice packet form **(via packet gateway, item 121 of Fig 1, col 2, lines 29-40)**; Huang et al disclosed that said second network device converts digital voice to analog voice stream form **(col 2, lines 53-67, col 3, lines 1-11, Figs 2-3)** but fails to disclose that said second network device converts voice between the analog voice stream form and the digital voice packet form when needed for routing voice between said first network device and said voice packet network. However, Oneill et al disclosed methods for converting voice forms from analog to digital and digital to analog forms in a network device **(item 618 of Fig 5, col 7, lines 42-65, Fig 5).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method for converting voice forms from analog to digital and digital to analog forms in a network device as taught by Oneill et al in the system of Huang to include in the said second network device to convert voice between the analog voice stream form and the digital voice packet form when needed for routing voice between said first network device and said voice packet network. One is motivated as such in order to use a digital to analog (D/A) and analog to digital (A/D) converter in the second network device to efficiently convert voice packets to route over the voice packet network.

Regarding claims 18, 39, Huang disclosed that said first network device is a telephone that captures, delivers, receives and reproduces voice in an analog voice stream form **(col 3, lines 3-11).**

Regarding claims 20-21,41-42, said voice packet network comprises an Internet switching network **(frame relay over wide area network and packet networks)** and wherein said second network **(item 112 of Fig 1)** device is an access device **(base station, item 115 of Fig 1, col 2, lines 3-40)**.

5. Claims 53 and 55-59 are rejected under 35 U.S.C. 103(a) as being unpatentable over Harrison et al [US Pat 5,796,727] over in view of Oneill et al [US Pat: 5,987,099].

Regarding claim 53, Harrison et al disclosed at least one circuit for use in a mobile communication device **(MS, items 39,40 of Fig 2)** for use in a communication network, the communication network operating to support voice and data communication **(PSTN, WAN, LAN of Fig 2)** and having at least a plurality of access devices **(item 41 of Fig 2)**, where each access device has a wireless transceiver **(cell controllers of Fig 2),** and the plurality of access devices use a polling protocol to manage wireless routing of data and voice packets among a plurality of mobile communication devices, the at least one circuit operational to **(Figs 2-3, col 3, lines 5-31, col 5, lines 8-40)**, at least: receive digital voice information converted from analog voice stream form and generate digital voice packets from the received digital voice information **(voice and data messages from**

wireless MTSO to LAN/WAN, col 4, lines 41-49, col 5, lines 7-51, Figs 2-3); receive digital voice packets, generate digital voice information from the received digital voice packets, and transmit the generated digital voice **(voice and data messages from LAN/WAN to wireless MTSO, col 5, lines 52-67, col 6, lines 46-58);** Harrison et al fails to positively disclose the feature for information for conversion to analog voice stream form for reproduction of voice; capture data and generate data packets from the captured data; and receive data packets and reproduce data from the received data packets. However, Oneill et al disclosed a method for said stationary network device **(base station controller, item 618 of Fig 5)** that converts analog voice streams received from said telephone into voice packets for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices **(col 5, lines 56-65, Fig 5).** Therefore it would have been obvious for one of the ordinary skills in the art at the time the invention was made to use the method of converting analog voice streams received from said telephone into voice packets at the stationary device for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices as taught by Oneill et al in the system of Harrison et al for the stationary network device to converts analog voice streams received from said telephone into voice packets for delivery via said hardwired and wireless networks to a selected one of said plurality of mobile network devices. One is motivated as such in order to convert analog voice streams to digital voice packets to efficiently route packets over the voice packet network for deliver to mobile network devices.

Regarding claim 55, Harrison et al disclosed wherein digital voice packets sent by the mobile communication device are selectively converted by one of the plurality of access devices in the communication network to analog voice stream form for delivery to a telephone that captures, delivers, receives and reproduces voice, **(col 6, lines 46-58);** and wherein the mobile communication device receives digital voice packets from the one of the plurality of access devices that selectively converts an analog voice stream received from the telephone to form the digital voice packets received by the mobile communication device **(communication between wireless mobile station and LAN, col 11, lines 14-47)**.

Regarding claims 56, Harrison et al disclosed where the one of the plurality of access devices stores incoming digital voice packets for an adjustable queuing time period before converting the digital voice packets into analog voice stream form **(stream manager for managing latency in stream queue, col 16, lines 7-19, Fig 12)** .

Regarding claim 57, Harrison et al disclosed wherein digital voice packets sent by the mobile communication device are selectively routed to a telephone switching network connected to the one of the plurality of access devices that converts the digital voice packets received from the mobile communication device into an analog voice stream form **(communication between wireless mobile station and LAN, col 11, lines 14-47)**, and wherein the one of the plurality of access devices selectively converts an analog voice stream received from the telephone switching

network into digital voice packets for routing to the mobile communication device **(col 11, lines 50-60);**.

Regarding claims 58, Harrison et al disclosed wherein the at least one circuit causes buffering of digital voice information for an adjustable queuing time period before converting the digital voice packets into analog voice stream form **(stream manager for managing latency in stream queue, col 15, lines 65-67, col 16, lines 1-19, Fig 12)** .

Regarding claim 59, Harrison et al disclosed wherein the at least one circuit directs one of the at least one access device to route digital voice packets via a user-selected communication network **(Figs 13, 14, col 16, lines 20—55).**

Response to Arguments

6. Applicant's arguments filed on 02/02/2009 have been fully considered and they are persuasive. Therefore the rejections communicated via previous office action has been withdrawn. However a new ground(s) of rejection has been made in view of a newly found reference.

Allowable Subject Matter

7. Claims 43-52 are allowed over prior art.

Claims 4, 8, 17, 25, 29, 38, 54 is objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims;

-Prior art fails to teach or render obvious the limitations of “where the voice packet network uses a polling protocol and incorporates spanning tree routing” as recited in claims 4, 8, 17, 25, 29, 38, 54.

Conclusion

8. Any inquiry concerning this communication or earlier communications should be directed to the attention to Venkatesh Haliyur whose phone number is 571-272-8616. The examiner can normally be reached on Monday-Friday from 9:00AM to 5:00 PM. If attempts to reach the examiner by telephone are unsuccessful, the examiner’s supervisor, Edan Orgad can be reached @ (571)-272-7884. Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the group receptionist whose telephone number is (571)-272-2600 or fax to 571-273-8300.

9. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, s. Should you have questions on access to

the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197(toll-free).

/Venkatesh Haliyur/

Examiner, Art Unit 2419

/Salman Ahmed/

Examiner, Art Unit 2419